IMPLEMENTATION OF AN ACTIVE COCHLEAR MODEL ON A TMS320C25

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ABSTRACT - This paper, which is the second of two papers describing the development of an active cochlear model submitted to this conference, outlines the implementation of the model on a Texas Instruments' TMS320C25 single-chip digital signal processor. The cochlear model contains both passive and active elements, in line with recent research findings in the field of auditory physiology. The passive system is operational at normal stimulus amplitudes, while the active system comes into play for low-amplitude stimuli. Tests of the implemented model have been carried out using sinewaves. This implementation could be used as a physiologically-based front-end processor for a speech recognition system.

INTRODUCTION

This paper describes the real-time implementation of an active cochlear model. The hardware consists of an analogue interface, a signal processing system based on the TMS320C25 digital signal processor (DSP), and an interface to an IBM PC to enable the PC to read the output of the DSP for plotting.

The auditory model used in this implementation consists of a cascade of digital filters, each of which has a different resonant frequency in the speech spectrum. It contains 22 sections covering the frequency range from 250 Hz to 3.5 KHz, and operates at a sampling frequency of 8 KHz. This model incorporates both passive and active elements. The passive system is operational at normal amplitudes, while the active system comes into effect at low amplitudes (Davis, 1983). Figure 1 shows a block diagram of one section of the auditory model. At each section of the model, the fluid pressure at the input to that section is converted into mechanical displacement. The pressure transfer function in the z-domain, for a single section, is given by (Ambikairajah et al., 1989; Linggard & Ambikairajah, 1986) :

$$\frac{V_o(z)}{V_i(z)} = K \frac{1 - a_0}{1 - a_0 z^{-1}} \frac{1 - b_1 + b_2}{1 - b_1 z^{-1} + b_2 z^{-2}} \frac{1 - a_1 z^{-1} + a_2 z^{-2}}{1 - a_1 + a_2} \qquad (1)$$

where V_i is the pressure input to the section, V_o is the pressure output from the section, a_0, a_1, a_2, b_1 and b_2 are digital filter coefficients and K is a gain factor. The membrane displacement transfer function is given by:

$$\frac{V_m(z)}{V_i(z)} = K \frac{1 - a_0}{1 - a_0 z^{-1}} \frac{(1 - b_1 + b_2)\, z^{-1}}{1 - b_1 z^{-1} + b_2 z^{-2}} \qquad (2)$$

where V_m is the membrane displacement.

The transduction of mechanical displacement to electrical energy takes place in the inner hair cells. The model of the inner hair cell used in the present work is a capacitor model, in which the input voltage corresponds to the spatially differentiated membrane displacement output of the auditory model. To detect the presence of a frequency component (or a formant when the stimulus is a speech signal), a decision algorithm is carried out on the inner hair cell outputs. If the amplitude of a component is below a certain threshold, then the corresponding section of the model is switched to the active (high-Q) state. This is accomplished by changing the digital filter coefficients for that section. Thus, implementing an active section requires no increase in computation over a passive section.

Extensive tests of the real-time model have been carried out using sinusoidal stimuli of varying frequency. Results of such tests are presented in this paper.

HARDWARE USED IN THE IMPLEMENTATION

Figure 2 shows a block diagram of the hardware used in the implementation of the active cochlear model. The hardware was constructed on a double-width wire-wrap board, and consists of three main components:

(1) an analogue-to-digital interface;
(2) a signal processing section based on the TMS320C25 fixed-point digital signal processor;
(3) an interface to an IBM PC which incorporates Static RAMs (SRAMs) which are used to pass results from the DSP to the PC.

It was decided to use a 12-bit parallel analog-to-digital converter (ADC), as opposed to a serial CODEC device, to eliminate the time necessary to convert the log-PCM output of the CODEC into the linear format required by the DSP. The ADC used is the Analog Devices AD ADC84-12, which has a conversion time of approximately 10 μs. This is connected to input port 0 of the DSP. The hardware is designed such that execution of an IN instruction by the DSP reads in the current latched input sample, and also sends a start conversion signal to the ADC (Troullinos & Bradley, 1987).

The DSP contains 544 16-bit words of internal RAM, which is used in this implementation to store coefficients and previous input/output values. For program space, external memory is used. To allow the system to operate at full speed, without wait states, the external memory must have an access time of less than 40 ns from address valid. The TMS27C291 (2Kx8) EPROM, which has an access time of 35 ns from address valid, is used. Since the instruction width of the DSP is 16 bits, parallel banks of EPROMs are used (the amount of program memory in the system is 4K words).

The third component of the system is the interface between the DSP and the PC. A digital input/output card is used to provide the interface between the PC bus and the SRAMs. At the end of each 16 msec processing frame, the DSP writes the output of the cochlear model to the SRAMs. The PC can then read these values from the SRAMs for storage, graphical display etc. To prevent bus contention, handshaking is carried out between the DSP and the PC using the DSP's BIO and XF pins. The SRAM used in this case is the MK41H68N-20, which has an access time of 20 ns from address valid.

Since the TMS320C25 uses fixed-point arithmetic, attention has to be paid to the issues of arithmetic overflow and scaling, e.g. in the case of a resonant section with a gain of greater than 1 at the resonant frequency. The TMS320C25 contains an on-board timer which can be programmed to generate a software interrupt after a certain number of machine cycles. Since the machine cycle time of the DSP is 100 ns, the timer is programmed to generate an interrupt after 1250 cycles, corresponding to a sampling interval of 125 μs.

This implementation of the active cochlea contains 22 digital filters. In order to implement an active cochlear model with more filters, a DSP with a shorter machine cycle time must be used. To minimise the number of data bus cycles, as much as possible of the data memory resides in the DSP's on-chip RAM. Since the program and data buses are multiplexed onto the same lines externally, instructions which involve an access to external data memory require an extra machine cycle for execution. This increases the number of machine cycles per filter, thus reducing the number of filters which can be implemented for a given sampling interval.

RESULTS OF THE REAL-TIME IMPLEMENTATION

The passive/active cochlear model was tested with sinusoidal stimuli. Figure 3(a) shows the membrane displacement output of the model 80 ms after the application of a sinewave of frequency 500 Hz to the input of the first filter. Figure 4(a) shows the corresponding inner hair cell output. The peak at filter number 14 is below the threshold level and, as a result, filter number 14 is switched to the active state (Figures 3(b) and 4(b)). When a filter is switched to the active state, it is allowed to remain in that state for a certain period of time, after which it is switched back to the passive state (see Figures 3(c) and 4(c)).

CONCLUSIONS

A real-time implementation of a model of the cochlea, incorporating both passive and active elements, has been described in this paper. Results obtained when the model was excited with sinusoidal stimuli were presented. The model is capable of providing increased amplification to a low-amplitude stimulus, by switching the appropriate filter to the active state. The results presented here are in good agreement with those obtained from a 22-filter model simulated on a PC using floating-point arithmetic.

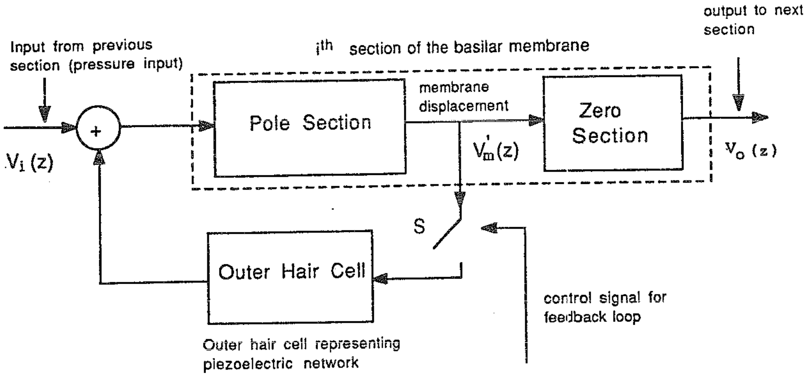

Figure 1. Block diagram of the ith section of the cochlear model.

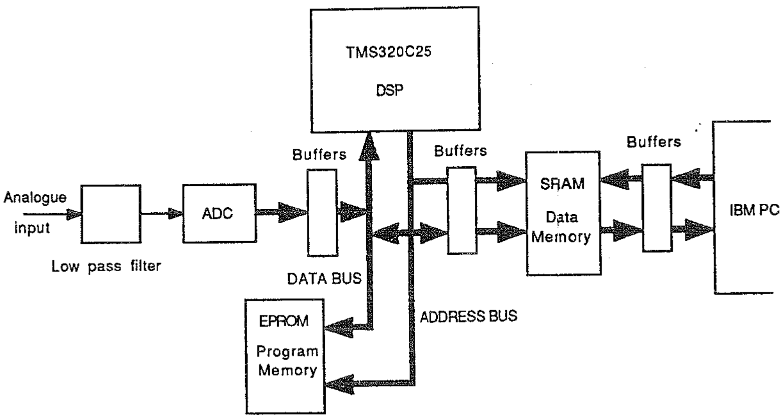

Figure 2. Block diagram of the hardware used in the real-time implementation of the cochlear model.

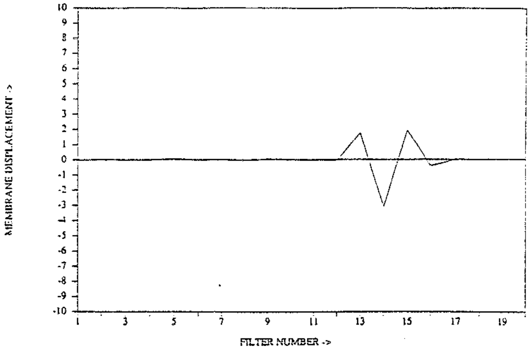

Figure 3(a) Membrane displacement 80 ms after application of a sinewave of frequency 500 Hz

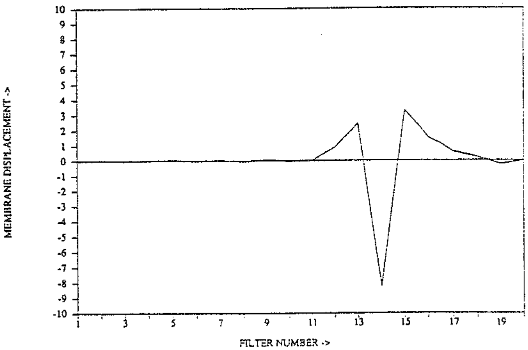

Figure 3(b) Membrane displacement after filter number 14 has switched to the active state

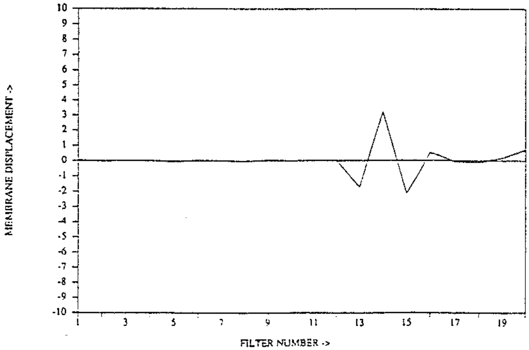

Figure 3(c) Membrane displacement after filter number 14 has switched back to the passive state

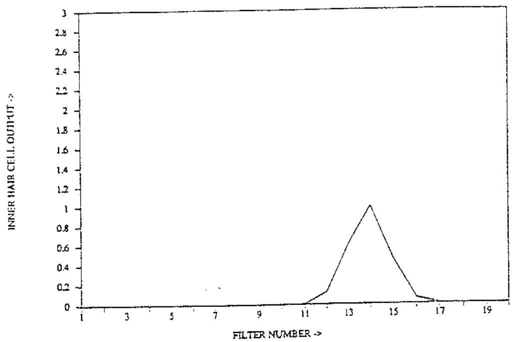

Figure 4(a) Inner Hair Cell output 80 ms after application of a sinewave of frequency 500 Hz

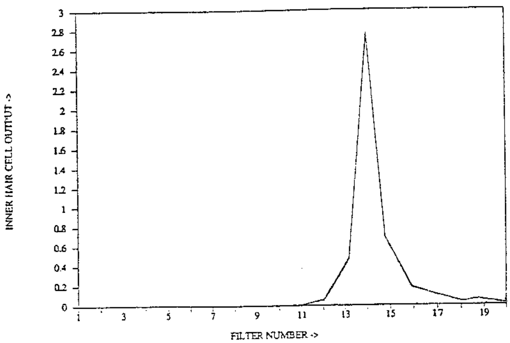

Figure 4(b) Inner Hair Cell output after filter number 14 has switched to the active state

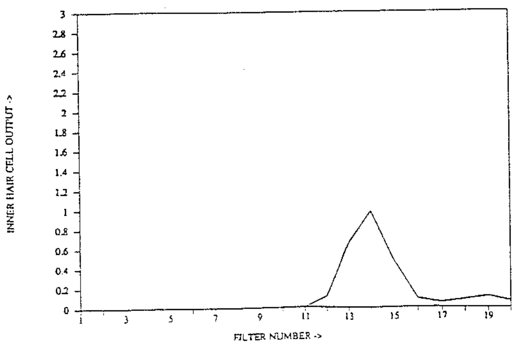

Figure 4(c) Inner Hair Cell output after filter number 14 has switched back to the passive state

ACKNOWLEDGEMENTS

This research was funded by a grant from EOLAS, the Irish Science and Technology Agency, under the Strategic Research Programme, 1987-1989.

REFERENCES

Ambikairajah, E., Black, N. & Linggard, R. (1989) *Digital filter simulation of the basilar membrane*, Computer Speech and Language **3**, 105-118.

Davis, H. (1983) *An active process in cochlear mechanics*, Hearing Research **9**, 79-90.

Linggard, R. & Ambikairajah, E. (1986) *A computational model of the basilar membrane.*, Proceedings of the Conference on Speech Science and Technology (SST-86), Canberra, 286-291.

Troullinos, G. & Bradley, J. (1987) *Hardware Interfacing to the TMS320C25*, Texas Instruments Application Report.